CLAIMS

1 – A method for processing an electric sound signal in which the following steps are implemented:

-an electric sound signal on the right (13) and an electric sound signal on the left (17) are processed to produce a processed electric sound signal on the right (53) and a processed electric sound signal on the left (62),

characterized in that to process

-the production of a first processed electric sound signal on the right (600) from the electric sound signal on the right (13) is simulated,

-the production of a second processed electric sound signal on the right (20) from the electric sound signal on the left (17) is simulated (603),

-the production of a third processed electric sound signal on the left (21) from the electric sound signal on the left (17) is simulated (602),

-the production of a fourth processed electric sound signal on the left (16) from the electric sound signal on the right (13) is simulated (601),

-a sound (63, 64) corresponding to these four processed electric sound signals is diffused.

2 – The method according to claim 1, characterized in that, to simulate

- a white acoustic sound signal on the right is produced (70) with an acoustic diffusion system (65), from a white noise electric signal (76),

- a corresponding acoustic signal received in the form of a modified white received electric sound signal on the right and a modified white electric sound signal on the left corresponding to the reception of the white acoustic sound signal on the right is detected with an acoustic detector (68, 69),

- a frequency spectrum on the right corresponding to a white noise electric signal on the right, and two received frequency spectrums are produced, respectively corresponding to the modified white received electric sound signal on the right and to the modified white received electric sound signal on the left,

- a first set of coefficients from frequency filters are produced from the frequency spectrum on the right and from the frequency spectrum of the modified white received electric sound signal on the right,

- a second set of coefficients from frequency filters are produced from the frequency spectrum on the right and from the frequency spectrum of the

modified white received electric sound signal on the left,

- a white acoustic sound signal on the left is produced (73) with an acoustic diffusion system (66), from a white noise electric signal (81),

- a corresponding acoustic signal received in the form of a modified white received electric sound signal on the left and a modified white electric sound signal on the right corresponding to the reception of the white acoustic sound signal on the left is detected with an acoustic detector (68, 69),

- a frequency spectrum on the left corresponding to a white noise electric signal on the left, and two received frequency spectrums are produced, respectively corresponding to the modified white received electric sound signal on the left and to the modified white received electric sound signal on the right,

- a third set of coefficients from frequency filters are produced from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the left,

- a fourth set of coefficients from frequency filters are produced from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the right,

- these four sets of coefficients form a quadrille of coefficient sets,

- and, to process, one filters the electric sound signals on the right and left with frequency filters whose parameters are given by this quadrille.

3 – The method according to claim 2, characterized in that

- the sets of coefficients are produced from the two spectrums by a component to component complex division of complex points from these components in each of these spectrums.

4 – The method according to one of claims 2 to 3, characterized in that, to diffuse

- the coefficients from four temporal filters (91-99) are produced from coefficients of the first, second, third and fourth frequency filters respectively.

5 – The method according to claim 4, characterized in that

- the coefficients of temporal filters are modified (195,196) by all or part of the following operations:

- normalization of temporal filters of a quadrille, on the maximum of the direct field or on quadratic average of the diffuse field,

- temporal reset (101) of the temporal filters with relation to each other,

- time lag of samples from a temporal filter,

- masking of some samples from the temporal filter (195, 196),

- alteration of amplitudes from certain samples from a temporal filter.

6 – The method according to one of claims 4 to 5, characterized in that

- in the coefficients from a temporal filter those whose rank is greater than a given rank are eliminated and where

- in the coefficients from a temporal filter those whose value is lower than a threshold (106, 107) are eliminated.

7 – The method according to one of claims 2 to 6, characterized in that

- quadrilles of sets of coefficients are produced for different configurations (301-305) of the acoustic diffusion system and or for different rooms (90,203) in which the acoustic diffusion system (83-85) is placed for the production of coefficients.

8 – The method according to claim 7, characterized in that

- one of the configurations is a configuration in cone of confusion (88, 89).

9 – The method according to one of claims 1 to 8, characterized in that, to diffuse

- the electric sound signals processed by the filters (26,31) are combined with the original unprocessed electric sound signals (13, 17),

- and a combined electric sound signal on the right and a combined electric sound signal on the left are obtained.

10 – The method according to claim 9, characterized in that, to combine

- a time lag is introduced between the acoustic electric sound signals processed by the filters and the original unprocessed electric sound signals.

11 – The method according to one of claims 9 to 10, characterized in that

- combined electric sound signals on the right and left are filtered on given frequency bands and,

- a delay is introduced in each of these frequency bands.

12 – The method according to claim 11, characterized in that

- combined electric sound signals on the right and left are filtered by using a high-pass filter, and

- high-frequency electric sound signals are obtained,

- combined electric sound signals on the right and left are filtered by using a low-pass filter, and

- low-frequency electric sound signals are obtained,

13 – The method according to claim 12, characterized in that

- a first delay is introduced in the low-frequency electric sound signals and

- a second delay is introduced in the high-frequency electric sound signals and

14 – The method according to claim 13, characterized in that

- the first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the left.

- the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the left.

15 – The method according to one of claims 1 to 14, characterized in that, to filter

- a signal transform of an electric sound signal is performed and a transformed signal is obtained,

- the transformed signal is multiplied by the filtering coefficients and a multiplied signal is obtained,

- the multiplied signal is transformed by an inverse transform,

- the filtering coefficients are coefficients of finite impulse response filters (118-121).

16 – The method according to claim 15, characterized in that, to perform the transform

- a frame of the electric sound symbol is divided into N blocks,

- the transform of each of the blocks is performed,

- the filtering coefficients are divided into N packets of coefficients,

- the N blocks of input data are multiplied two by two by the N packets of filter coefficients, and

- the multiplied blocks are added to obtain the multiplied signal.

17 – The method according to claim 16, characterized in that to divide

the frame and to calculate the transform

- the transform of each of the N blocks is calculated successively, and

- the transformed blocks are transmitted to a delay line at N outputs.

18 – The method according to one of claims 16 to 17, characterized in that, to divide the frame into N blocks

- an electric sound signal is stored in a circular buffer memory with capacity proportional to the nth of the frame of the electric sound signal.

19 – The method according to one of claims 16 to 18, characterized in that

- to divide a frame of the signal into N blocks, double blocks are formed that are overlayed on each other by half,

- the transform of each of the double blocks is performed,

- the N packets of coefficients are completed by the constant samples to obtain double packets,

- each of the N double blocks are multiplied by one of the N double packets and multiplied double blocks are obtained, and

- the multiplied blocks are extracted from the multiplied double blocks.

20 – The method according to one of claims 1 to 19, characterized in that, to simulate

- an artificial head that comprises two acoustic detectors (68,69) is placed in a median axis of two acoustic diffusion systems (65,66),

- an electric signal in the form of a Dirac comb is applied simultaneously as input to the two acoustic diffusion systems,

- these direct fields and these crossed fields received by the acoustic detectors are aligned two by two by varying the position of the artificial head.

21 – The method according to one of claims 1 to 20, characterized in that, to diffuse

- equalization functions are incorporated in the cells situated upstream from the Fourier transform cells.

22 – The method according to claim 21, characterized in that

- the frequency components of four frequency filters obtained from four modified temporal filters are adjusted independently.

23 – The method according to one of claims 1 to 22, characterized in that, to diffuse

- the phase and/or the amplitude of the temporal filter coefficients (91-99) are modified along all or part of the impulse response.

24 – The method according to claim 15, characterized in that, to perform the transform

- the filtering temporal coefficients are divided into Q slots (HDD1-HDD4) of coefficients with progressive length M, 2M, 4M,...(2^(Q-1))M points.

- the transform of each of these slots is performed and transformed slots are obtained,

- a frame of the electric sound signal is divided into blocks (x1-x8) with a length of M points,

- the transform of each of these blocks is performed and transformed blocks are obtained,

- the transformed blocks are multiplied by the transformed slots and corresponding multiplied blocks are obtained by inverse transformation to the blocks of signals that half-overlap each other two by two in time.

25 – The method according to claim 24 characterized in that to perform the inverse transformations of multiplied blocks

- a first multiplied block (618) with a length of 2P x M points, a temporal block (613) corresponding in time to this first multiplied block, a second multiplied block corresponding in time to a second temporal block are modulated (632), this first and second temporal block are overlayed by half in time, and

- a modulated block (620) with a length of 2P x M points is obtained, then

- this modulated block with a length of 2P x M points is added (633) to the second block, and

- a combined block (621) with a length of 2P x M points is obtained.

26 – The method according to claim 25, characterized in that, to modulate

- the odd components of a multiplied block with a length of 2M points wherein the block corresponding to it in time is overlayed with another is multiplied by –1, and the even components are multiplied by +1.

27 – The method according to one of claims 25 to 26 characterized in that to perform the inverse transformations of multiplied blocks with a length

of 2M points

- the even components of the combined block with a length of 2P x M points are selected (604) and

- an even block with a length of 2(P-1) x M points is obtained

- this even block is multiplied by 1/2 and the result of this multiplication is added (607) to an auxiliary multiplied block with a length of 2(P-1) x M points and

- a compensation block (623) is obtained.

27 – The method according to one of claims 25 to 26 characterized in that to perform the inverse transformations of multiplied blocks with a size of (2P)M,

- the odd components of the combined block with a size of 2P x M points are selected (605) and

- an odd block (624) with a length of 2(P-1) x M points is obtained,

- an inverse transform (606) of this odd block with a length of (2(P-1))M points is performed and

- an odd inversed block (625) is obtained that is situated in the temporal domain, then

- this odd inversed block (625) is multiplied (606) by a complex coefficient conjugated from a complex coefficient W(n) and

- an odd normalized inversed block (626) with a length of 2(P-1) x M points is obtained.